IN THE CLAIMS

The claims are repeated without amendment and for the Examiner's convenience as follows:

1. (Previously Presented) A hearing aid, comprising:

a microphone to receive an input signal;

and

a digital processor to process the input signal at a gain, wherein the processor includes an inhibitor to inhibit distortions and an adjuster to adjust the gain of the input signal, wherein the inhibitor smoothes an envelope of the input signal so as to inhibit distortions arising from apparent modulation of the input signal due to sampling of the input signal.

2. (Original) The hearing aid of claim 1, wherein the inhibitor creates two representations that are orthogonal to each other in phase.

3. (Original) The hearing aid of claim 1, wherein the inhibitor includes a multiple of time-constant circuits to smooth the envelope of the input signal.

4. (Original) The hearing aid of claim 1, wherein the inhibitor includes a detector having a Hilbert filter so as to smooth the envelope of the input signal.

5. (Original) The hearing aid of claim 1, wherein the inhibitor includes an estimator that estimates at least one of a minimum and a maximum of two representations of the input signal that are orthogonal to each other in phase, wherein the estimator allows a linear extraction of the amplitude so as to smooth the envelope of the input signal.

6. (Previously Presented) A method comprising:

sampling an input signal;

smoothing an envelope of the input signal; and

adjusting the gain if the envelope is greater than a threshold, wherein the smoothing inhibits distortions arising from apparent modulation of the input signal produced by sampling the input signal.

7. (Original) The method of claim 6, wherein smoothing includes creating two representations of the input signal, wherein the two representations are orthogonal to each other in phase.

8. (Original) The method of claim 7, wherein creating includes creating the magnitude of the two representations to approximate the magnitude of the input signal.

9. (Original) The method of claim 7, wherein smoothing includes smoothing using a Hilbert filter.

10. (Original) The method of claim 9, wherein smoothing includes squaring each sample to form a squared sample, summing each squared sample with other squared samples to form a sum, and taking a square root of the sum.

11. (Previously Presented) An apparatus for processing a digital audio signal, comprising:

an adjuster to adjust amplification of the digital audio signal; and

a detector to form a smooth envelope that is a rectified version of the digital audio signal, wherein the detector presents the smooth envelope to the adjuster, and wherein the smooth envelope excludes apparent modulation of the digital audio signal.

12. (Previously Presented) The apparatus of claim 11, further comprising a preamplifier to amplify the input signal, wherein the adjuster adjusts amplification of the preamplifier.

13. (Previously Presented) The apparatus of claim 12, further comprising an analog-to-digital converter that receives the input signal, which is amplified by the preamplifier, and produces a digitized input signal.

14. (Previously Presented) The apparatus of claim 13, further comprising a filter to receive the digitized input signal and to produce a filtered input signal that excludes a direct-current component of the digitized input signal.

15. (Previously Presented) The apparatus of claim 14, further comprising a digital-to-analog converter that receives a digital adjustment from the adjuster, produces an analog adjustment, and presents the analog adjustment to the preamplifier.

16. (Previously Presented) A hearing aid for processing an input signal, comprising:

a preamplifier having a gain to amplify the input signal and produce an amplified input signal;

a sampler to sample the amplified input signal;

a detector to form a smooth envelope that is rectified; and

an adjuster to adjust the gain of the preamplifier if the smooth envelope is greater than a threshold to reduce distortions due to an apparent modulation arising from sampling of the amplified input signal.

17. (Original) The hearing aid of claim 16, further comprising a filter to produce a filtered input signal that excludes direct current.

18. (Original) The hearing aid of claim 17, wherein the detector includes a Hilbert filter, wherein the Hilbert filter receives the filtered input signal, and produces two signals that are 90 degrees out of phase with each other.

19. (Original) The hearing aid of claim 18, wherein the detector squares each signal of the two signals, sums the two squared signals to form a sum, and takes the square root of the sum to form the smooth envelope of the input signal.

20. (Original) The hearing aid of claim 18, wherein the detector squares each signal of the two signals and sums the two squared signals to form the smooth envelope of the input signal.

21. (Previously Presented) A gain control for processing an input signal, comprising:
    a sampler to sample the input signal;
    a detector to detect an envelope of the input signal;
    an adder to provide a difference between the envelope and a threshold; and
    an adjuster that adjusts a gain based on the difference,
    wherein the detector is adapted to reduce apparent modulation arising from sampling of the input signal.

22. (Previously Presented) The gain control of claim 21, further comprising a filter that removes low frequencies, wherein the filter receives the input signal, removes frequencies less than about 100 Hertz from the input signal, and presents the input signal to the detector.

23. (Previously Presented) The gain control of claim 22, further comprising a digital delay element that receives the input signal and presents a delayed input signal.

24. (Previously Presented) The gain control of claim 23, further comprising a first filter and a second filter in a Hilbert filter arrangement, wherein the first filter receives the delayed input signal and filters the delayed input signal to form the first filtered input signal, and wherein the second filter receives the input signal and filters the input signal to form the second filtered input signal.

25. (Previously Presented) The gain control of claim 24, further comprising a first multiplier and a second multiplier, wherein the first multiplier receives the first filtered input signal and squares the first filtered input signal to form a first squared signal, and wherein the second multiplier receives the second filtered input signal and squares the second filtered input signal to form a second squared signal.

26. (Previously Presented) The gain control of claim 25, further comprising another adder to add the first squared signal and the second squared signal to form a sum-of-square signal.

27. (Previously Presented) The gain control of claim 26, further comprising a limiter that receives the sum-of-square signal, limits the sum-of-square signal to a desired range, and presents a limited signal to the adder that provides the difference between the envelope and the threshold.

28. (Previously Presented) A gain control operating on an input signal, comprising:

a sampler for sampling the input signal;

a detector to detect an envelope of an input signal using a Hilbert filter arrangement;

an adder to provide a difference between the envelope and a threshold; and

an adjuster that receives the difference, a release time constant, and an attack time constant, wherein the adjuster adjusts a gain if the difference is one of two conditions, wherein the two conditions includes being a negative number and being a positive number, wherein the adjuster increases the gain if the difference is negative, and wherein the adjuster decreases the gain if the difference is positive,

wherein the detector is adapted to reduce apparent modulation arising from sampling of the input signal.

29. (Previously Presented) The gain control of claim 28, wherein the adjuster receives a previous gain, wherein if the difference is negative, the adjuster increases the gain by shifting the bits of the previous gain to the right by the release time constant to form a new gain and taking the negative of the new gain.

30. (Previously Presented) The gain control of claim 29, wherein if the difference is positive, the adjuster decreases the gain by shifting the bits of the difference to the right by the attack time constant to form the new gain.

31. (Previously Presented) The gain control of claim 30, further comprising a width adjuster that adjusts the word with of the previous gain and presents an adjusted previous gain.

32. (Previously Presented) The gain control of claim 31, further comprising another adder that adds the new gain and the adjusted previous gain to form the gain.

33. (Previously Presented) The gain control of claim 32, further comprising a limiter to the limit the range of the gain so that the gain is positive.

34. (Previously Presented) The gain control of claim 33, further comprising a buffer that stores the gain and presents the stored gain, wherein the stored gain is defined as the previous gain, which is presented to the adjuster and the width adjuster.

35. (Previously Presented) The gain control of claim 34, further comprising a rounding circuit that rounds the stored gain to a smaller precision value so as to be compatible with the input width of subsequent circuitry that includes a digital-to-analog converter.

36. (Previously Presented) A gain control operating on an input signal, comprising:

a filter to block low frequencies of the input signal;

a sampler to sample the input signal;

a detector to detect an envelope of the sampled input signal using a Hilbert filter arrangement;

an adder to provide a difference between the envelope and a threshold; and

an adjuster that receives the difference, a release time constant, and an attack time constant, wherein the adjuster adjusts a gain if the difference is one of two conditions, wherein the two conditions includes being a negative number and being a positive number, wherein the adjuster increases the gain if the difference is negative, and wherein the adjuster decreases the gain if the difference is positive, wherein the detector is adapted to reduce the apparent modulation arising from sampling of the input signal.

37. (Previously Presented) The gain control of claim 36, wherein the filter to block low frequencies includes a first digital delay that receives the input signal and presents a delayed input signal.

38. (Previously Presented) The gain control of claim 37, wherein the filter to block low frequencies includes a first adder that determines a difference between the input signal and the delayed input signal.

39. (Previously Presented) The gain control of claim 38, wherein the filter to block low frequencies includes a first multiplier that multiplies the difference and a scale to form a scaled signal, wherein the scaled signal inhibits the filter from overflow.

40. (Previously Presented) The gain control of claim 39, wherein the filter to block low frequencies includes a second adder that adds the scaled signal and a blocked signal to form a filtered signal.

41. (Previously Presented) The gain control of claim 40, wherein the filter to block low frequencies includes a second digital delay that receives the filtered signal and presents a filtered signal that is delayed.

42. (Previously Presented) The gain control of claim 41, wherein the filter to block low frequencies includes a second multiplier that multiplies the filtered signal that is delayed and an alpha signal to form a blocked signal, wherein the alpha signal determines a range of frequencies that will be blocked by the filter.

43. (Previously Presented) A gain control for processing an input signal, comprising:

a sampler to sample the input signal;

a detector to detect an envelope of the input signal using IIR filters in a Hilbert filter configuration;

an adder to provide a difference between the envelope and a threshold; and

an adjuster that receives the difference, a release time constant, and an attack time constant, wherein the adjuster adjusts a gain if the difference is one of two conditions, wherein the two conditions includes being a negative number and being a positive number, wherein the adjuster increases the gain if the difference is negative, and wherein the adjuster decreases the gain if the difference is positive,

wherein the detector is adapted to reduce apparent modulation arising from sampling of the input signal.

44. (Canceled)

45. (Previously Presented) The gain control of claim 43, wherein each IIR filter includes a first delay, a second delay, and a scale element, wherein the input signal is delayed by the first delay, delayed by the second delay, and scaled by the scale element to form a scaled signal.

46. (Previously Presented) The gain control of claim 45, wherein each IIR filter includes a first adder that determines a difference between the input signal and a feedback signal.

47. (Previously Presented) The gain control of claim 46, wherein each IIR filter includes a multiplier that multiplies the difference and a beta signal to form a modified signal, wherein the beta signal modifies the phase of the difference.

48. (Previously Presented) The gain control of claim 47, wherein each IIR filter includes a third delay that delays the modified signal to form a filtered signal.

49. (Previously Presented) The gain control of claim 48, wherein each IIR filter includes a fourth delay that delays the filtered signal to form the feedback signal.

50. (Previously Presented) A method for controlling a gain of an amplifier, comprising:
blocking low frequencies from an input signal that is digitized;
forming an envelope that lacks apparent modulation using at least one Hilbert filter; and
subtracting the envelope from a threshold to form a difference, wherein the difference is used to control the gain.

51. (Original) The method of claim 50, wherein blocking includes blocking low frequencies that are less than about 100 Hertz.

52. (Original) The method of claim 50, further comprising determining if the difference is greater than zero.

53. (Original) The method of claim 52, further comprising shifting the bits of the difference to the right by an attack constant to form a decreased gain.

54. (Original) The method of claim 53, further comprising shifting the bits of a negated signal to the right by a release constant to form an increased gain.

55. (Original) The method of claim 54, further comprising switching for presenting the decreased gain as the gain if the difference is greater than zero, or else the act of switching presents the increased gain as the gain if the difference is less than zero.

56. (Original) The method of claim 55, further comprising summing the gain and the feedback signal that is delayed to form a modified gain signal.

57. (Original) The method of claim 56, further comprising presenting a final gain to an analog-to-digital converter, wherein the final gain is zero if the modified gain signal is less than or equal to zero, and wherein the final gain is one if the modified gain signal is greater than one.

58. (Original) The method of claim 57, further comprising delaying the final gain to produce the feedback signal that is delayed.

59. (Original) The method of claim 58, further comprising negating the feedback signal that is delayed to form the negated signal.